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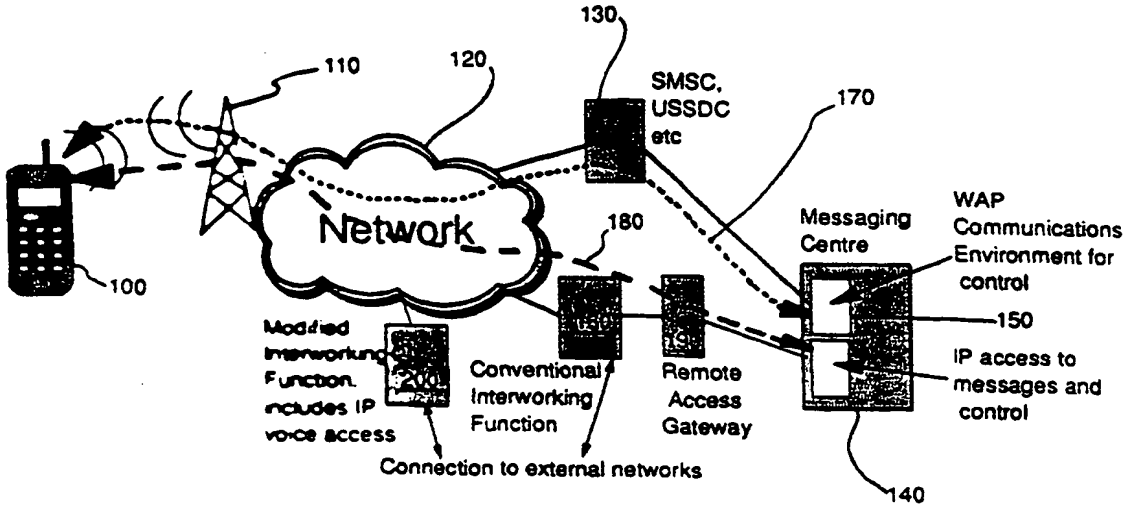

(57) Abstract

A problem in the mobile communications field is that enhanced voice services such as envisaged by the WAP Forum for mobile users of wireless networks would require use of two bearer services to deliver said services. This is inefficient and potentially more costly to the user and network operator. The problem is solved by use of a voice over IP technology and with digitising and packetizing of voice data and control data and transmission of the discrete packets over a single circuit-switched connection using IP based data communication to provide both the IP Voice connection and associated control for enhanced voice services.

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CONCURRENT IP BASED VOICE AND DATA SERVICES VIA WIRELESS NETWORKS

Field of Invention

The present invention relates to communication of audio information in a wireless communications network - that is, a network having wireless links between some communication devices (such as mobile telephones, communications-centric PDAs or communications-enabled computers) and network access nodes (commonly referred to as base stations in cellular networks). Each network access node receives information from and transmits information to local wireless devices.

Background

Today's advanced wireless networks, especially the digital cellular networks such as the Global System for Mobile communication (GSM) and Code Division Multiple Access (CDMA) networks, provide a variety of bearer services available to the user, each with different characteristics and costs of use. While such networks differ in many aspects, the present invention is applicable to all of them and so can be described with reference to any one - for example GSM.

GSM provides timeslots for both control information and users' calls within its 8 time slots per radio channel architecture which is a variation of Time Division Multiple Access (TDMA). A channel in this context is a particular operating frequency and parameters defining an end-to-end transmission path at that frequency. Each time slot provides a nominal capacity of 22.8kbps which includes the necessary channel coding, which results in a nominal 13kbps for voice services and 12kbps for the fastest data service. This latter 12kbps data rate is further reduced to a nominal 9600bps by use of the Radio Link Protocol which provides extra data error correction.

The so called "control channel" (which in GSM is one of the eight time slots per channel, rather than a truly separate channel) is used by mobiles and the network for call control, namely registering the users active presence on the network when first turning on the telephone or entering an area of network coverage, making or answering a call. Any spare control channel capacity is used for low rate data services. Two low speed or narrow band data services are known as SMS (Short Messaging Service) or Unstructured Supplementary Service Data (USSD). The SMS and USSD data services are, to all intents, low speed packet data services, since the messages can be sent whenever the network has the spare capacity to deliver them. The voice and high speed data services using a whole

timeslot are known as circuit-switched services since a circuit is setup from the user to the end-point, which may be another telephone in the case of a voice call or another computer if a data call.

Users have typically used the voice services without particular knowledge or care about the complexity of service provision. They simply use the keypad to enter the desired telephone number and press the appropriate PROCEED key, the telephone and network taking care of the rest. Users often receive SMS messages from the network's voice response systems in response to other users leaving messages when they have been unable to get answers to their calls. More recently SMS, USSD and equivalent services have been used to deliver information such as traffic information to users who subscribe to such services. However, if users have wanted to send messages they either needed a computer to interface to the SMS or USSD access of the phone or use complex sequences of keystrokes to generate and send such messages using the basic phone functions. If a user wants to use the circuit switched data service then an integrated or attached computer is necessary to handle the data applications.

Many data applications have been developed and used via GSM and other networks using conventional Internet-based and other communications, but the Internet Protocols (TCP/IP and UDP/IP) are becoming the defacto standard. These are far from optimal for this use at the present time.

The Wireless Application Protocol (WAP) Forum is an industry forum with an aim of delivering advanced telephony and information services to users of mobile wireless devices such as phones, pagers, smart phones and personal digital assistants. The WAP Forum has produced a set of specifications to meet these aims and continues to complete and enhance this task. The fundamental concept of WAP is to deliver services using internet-based technology, the user interacting with the phone and associated services using a micro-browser, the information being delivered by communication protocols similar to those of the Internet's Internet Protocol (IP) and HyperText Transfer Protocol (HTTP). The WAP protocols are known as WDP (Wireless Datagram Protocol) which is equivalent to the Internet's UDP/IP, WTP (Wireless Transaction Protocol) which provides acknowledged delivery and optionally segmentation and reassembly, and WSP (Wireless Session Protocol) which is similar to HTTP.

There is additionally WTLS (Wireless Transport Layer Security) to deliver authentication and secure data delivery between WAP client and WAP proxy (a proxy is a server computer acting as an intermediary between the client and the communications network, and in particular the WAP proxy is responsible for delivery of data content in a form which is independent of the network's data communications protocol). The communications protocols

have been designed to operate over the bearer services available in today's networks, such as the narrow band SMS and USSD data services as well as the higher data rate circuit-switched data services at 9600bps or lower. Content is in the form of Wireless Markup Language (WML) and WMLScript, which are based on the Internet's eXtended Markup Language (XML), and HyperText Markup Language (HTML), and JavaScript respectively (though WML is both a subset and superset of HTML and WMLScript is a subset and superset of JavaScript). Thus applications and services can be provided in WML or WMLScript without having to worry about which bearer service is being used - such as SMS, USSD, or Circuit Switched. WML content is in the form of decks of cards (collections of one or more 'cards', each comprising a fully specified piece of WML or WMLScript content or function).

The following description of the present invention will refer to WAP by way of example, but a similar approach could be adopted using conventional TCP/IP or UDP/IP communications over an HTTP session and an HTML, XML or XMLScript based application.

Now consider an enhanced voice based service, such as a voicemail service as provided by IBM Corporation's DirectTalk products. When voice messages arrive in the user's intelligent voice message service, the messages are recorded and via some method irrelevant to this discussion the callers' identities are obtained (calling line ID or voice recognition are two options for this identification). The message service generates the appropriate WML to describe the various callers' identities and call control options (for example 'listen', 'discard', 'save', 'forward'), and sends it to the user either when solicited (PULL model) or unsolicited (PUSH model). The delivery could be via any of the bearer services available to both the WAP client and the WAP proxy, which has the responsibility of delivering content to users securely, efficiently and reliably. The use of SMS or USSD has advantages since they can be used while a circuit switched voice call is in progress and they do not require a call to be set up. If the user responds to this message with call options requiring a voice call to be set up then the WAP based phone has all the capabilities to establish the call using the WML and internal supporting WAP libraries. At any time the user can use the telephone's micro browser interface to effect changes in the function of the service required (STOP, REPEAT, etc) and to convey such command again via the established SMS or USSD service.

To enable the user to work with an in-progress voice call, it is necessary for the WAP-enabled phone to implement an interface to the DTMF function (Dual Tone Multi Frequency signals are the signals generated by pressing a telephone's touch keys in a "Touchtone" phone) or some other

signalling level function, the availability of which might be very dependent on the network type being used or even the facilities provided even if the network architecture catered for it.

If SMS, USSD or an equivalent service is being used for signalling while the voice call is in progress then the user will be paying for both services - the voice call connection and the SMS or equivalent service used for control signals.

US-A-5799251 discloses a potential problem in radio telephone systems in that Short Data Messages which are sent on the control channel among control signals can block the control channel, causing interference in the control signalling and potentially affecting the speech traffic. A solution is then proposed which involves reserving a radio channel specifically for transmission of user's data messages, this reserved channel operating like a second control channel. A problem with this solution is network resource usage, and associated cost to the user, since the user requires an additional channel.

US-A-5790551 discloses dynamic assignment of an available packet data traffic channel for transmission of packetized data based on which channels will be free for a specified time period, the dynamically assigned channel being separate from the data control channel. Thus, no dedicated channel is required for data transmission, permitting more efficient and flexible use of available communication channels.

Summary of Invention

According to a first aspect of the invention, there is provided a method for use in the communication of audio information between a wireless communication device and a remote communication device via a wireless communication network, the method comprising: at one of said communication devices, providing audio information in digital form in discrete packets and providing call control information in digital form in discrete packets; establishing a call connection between the wireless communication device and the remote communication device; and transmitting both the discrete audio packets and the discrete call control packets via said single call connection.

Transmitting audio information (voice or synthesized voice data) and control information via a single logical end-to-end connection avoids the need for a separate connection to be maintained throughout the call for control information or for a separate connection to be established at numerous times within the call. The invention avoids the need to use two bearers throughout each call for each direction of communication between

the communication devices and this reduces the cost of the user's call (in particular, by avoiding the need for two connections between a mobile device and a network access node) and increases the available network capacity for other users. That is, by removing the need for concurrent use of SMS or USSD low-bandwidth or narrow band bearer services or a separate higher rate bearer to convey control information for the call, the bearer's capacity may be utilised for other SMS and USSD message services or to make additional capacity available for voice traffic and its associated controls. Furthermore, reducing the demand for SMS or USSD type bearer services to manage ongoing control of services reduces the likelihood of needing to dynamically assign additional low rate data service capacity.

As well as using less network resource than prior art solutions, the present invention has additional advantages. Firstly, each call drains the telephone's battery power more slowly if the invention is used, because only one connection is maintained after call set-up. Secondly, in embodiments of the invention in which call control information is sent in packets over a circuit switched call connection, the latency is both lower and more predictable than the latency of USSD or SMS messages. For interactive voice applications, the long latency of SMS is unacceptable; USSD is faster, but nevertheless has a latency which is longer and less predictable than a circuit switched connection. Ensuring that voice data and control data have the same latency is, of itself, beneficial since it avoids voice data having to be held to await control information. For example, if a user selects 'FORWARD' to next message, they want to be moved to the next message without delay.

The invention goes against the general teaching of the prior art which has dedicated channels or timeslots for voice data and for control data and which uses for each of these data types a communications protocol which has been optimised over time for the respective data type. In other words, the focus of the prior art has been on optimising voice processing and associated transmission protocols and otherwise enhancing the voice services within the constraints of the available data capacity of the respective channel architecture standards, without focusing on how to reduce the cost of voice services. This can be understood in the context of the available data communication capacities which have in the past required considerable development effort merely to keep pace with conflicting demands for high voice quality, comprehensive error correction, etc.

According to a preferred embodiment of the invention, the step of establishing a call connection follows the user of a calling device (for example, a mobile telephone) requesting access to a messaging centre for

use of interactive messaging services, which in turn follows either a PUSH notification of new messages or a regular user or mobile device initiated check (content PULL). Establishing the connection preferably comprises sending an initial notification (for example, via SMS or USSD) from the calling device to a called device (for example, the messaging centre), informing the called device of the calling device's intention to establish an interactive voice oriented session, and then the called device and caller device establishing a connection (possibly after sending an acknowledgement to the caller). Upon establishment of this connection, and verification of identities, the devices notify each other that the new channel will be used for subsequent transfers of control information. An alternative and simpler method which involves the same steps except that it omits the initial notification stage is equally acceptable.

After the connection has been set up, only a single call connection is required for transmission of both voice data and associated ongoing call control information. The voice data packets and control packets are transmitted using a common transport protocol. This may be, for example, a protocol based on current WAP protocols over IP using WML or WMLScript as data content formats. At the end of the interactive session, the calling device may indicate the end of the session or simply cease the connection, one of these actions being predefined as the condition for resumption of the use of the shared low bandwidth bearers for subsequent notification and access.

Voice communication requires regular delivery of content, but humans are also tolerant of minor disturbances in speech such as clicks and 'holes' where a small amount of information is not delivered. The invention preferably transmits speech content and control information as discrete packets via a single circuit-switched connection, allowing regular delivery to be achieved with low overhead and complexity. A similar approach over packet networks requires a little more control over regular delivery and contention, to achieve error and packet loss resilience, including some buffer control for voice while the control information is small and can fill the gaps in buffer managed time.

The invention preferably uses 'voice over IP', which entails digitising the voice information and sending it across the network in discrete packets using the Internet Protocol (in contrast to the traditional circuit-switched protocols of the telephone networks, but nevertheless preferably using a circuit switched connection). However, the invention is not limited to any particular implementation of 'voice over IP' and includes any means of conveying suitably encoded voice over an IP-based connection.

In network architectures such as GSM which divide their communications channels into multiple timeslots, the invention provides most benefits if combined with an appropriate coding of voice information which leaves spare capacity within a single time slot, or alternatively using a non-standard data transmission rate which provides spare capacity, such that voice and control information can be provided within the capacity of a single timeslot.

In a second aspect of the invention, there is provided a wireless communication device (such as a mobile telephone, or a PDA with communications facilities) including: means for encoding audio information in a digital form in discrete packets and means for generating discrete packets of digital call control information; means for establishing a call connection via an access node of a communications network; means for transmitting both the discrete audio packets and the discrete control packets to the local access node via said call connection; means for receiving discrete packets containing audio information and discrete packets containing call control information from the network access node via said call connection; and means for decoding received audio information packets and control information packets.

A third aspect of the invention provides a messaging centre for providing voice-based communications services in a communications network supporting wireless communications, the network including access nodes (or base stations) for receiving signals from, and for transmitting signals to, mobile telephones within their local cell, the messaging centre including: means for encoding audio information in a digital form in discrete packets and means for generating discrete packets of digital call control information; means for establishing a call connection between the messaging centre and a mobile telephone within the network; means for transmitting the encoded discrete audio packets and discrete control packets to the mobile telephone via said call connection; means for receiving discrete packets containing audio information and discrete packets containing call control information sent across the network from the mobile telephone; and means for decoding received audio information packets and control information packets.

The messaging centre preferably includes means for transmitting to a wireless communications device via a wireless communications network a first message including, within the message content, one or more menus of selectable operations, and means responsive to receipt of a message, sent from said wireless communications device in response to the first message (either with or without user interaction), for establishing said call connection for provision of a selected operation. The messaging centre can thus provide a unified messaging service to deliver a number of different

data content types (e-mail, voice, e-commerce transactions, database lookup operations) with the requests made using selectable menus delivered as WML, WMLScript or some other equivalent language, and the response to such actions can be converted to a voice format using text-to-speech technology for example.

Brief Description of Drawings

A preferred embodiment of the present invention will now be described in more detail, by way of example, with reference to the accompanying drawings in which:

Figure 1 is a schematic representation of a data communications network according to the prior art; and

Figure 2 is a schematic representation of a data communications network implementing the invention according to a preferred embodiment.

Description of Preferred Embodiment

A wireless communications network (i.e. a network including wireless links) according to the prior art is shown schematically in Figure 1. A mobile telephone 10 is shown communicating with a remote Messaging Centre 50, via two different transmission paths 70,80. The telephone communicates with network 30 via a network access node 20. The Messaging Centre is an example of a remote communication device to and from which voice data is sent and received, which provides voice services such as voicemail recording when a user's telephone is not contactable (e.g. switched off, on another call, or out of range of a network access unit). Communications between the mobile telephone and either the Messaging Centre or another communications device involves a first connection being established for transmission of control information in the form of SMS or USSD alert messages via a Short Message Service Centre (SMSC) or Unstructured Supplementary Services Data Centre (USSDC) 40. A second connection via a different transmission path is then established in response to the alerts, and this is used for the communication of voice data between the mobile telephone and the Messaging Service or another communications device. Communications with the Messaging service is via a conventional Interworking Function (IWF) 60 which converts between the data transfer protocols of fixed telephone lines and of wireless communications. As well as being used for the alert messages to trigger establishment of the second connection for transfer of voice data, this first connection will be used for subsequent transfers of call control information. A WAP communications environment 90 provides control data access to various

bearers including SMS, USSD, etc. Conventional voice access (e.g. PSTN or ISDN) is used for conventional access to messaging services.

This conventional arrangement has the problem that two separate connections must be maintained throughout the call, even if there is spare capacity on each connection. This use of network resources is typically charged back to the user.

Figure 2 shows a communications network suitable for implementing an embodiment of the present invention, in which a first control path 170 is used for transfer of alerts between a mobile telephone 100 and a Messaging Centre 140 via a Short Messaging Service Centre or Unstructured Supplementary Services Data Centre 130. After an initial alert has been received, or after a SMS or USSD response message has been returned, a separate call connection 180 is established between the mobile telephone and Messaging Centre via a Remote Access Server or Gateway 190 (described below). The messaging centre is assumed to have a suitable data communications environment for the delivery of notifications, menus to control any options to manage the messages, etc, and this is assumed to be WAP based.

Having established this separate connection, all subsequent session control information as well as voice data is sent via this connection and the connection used for transfer of alerts is no longer required. (Of course, if the call connection terminates during the call, then a new connection will have to be established and the communications path may be different on reconnection - for example the mobile telephone may have moved to a new cell of a cellular network. However this should not cause any problems as the new connection would be accompanied by a new identity (IP address) notification for the mobile telephone).

The transmission of voice and control data via a single connection involves processing voice signals (digitizing, encoding including compressing, and packetizing the audio data) such that the voice and control data can be carried over a common communication protocol. The replacement of the initial control channel by the single voice and control connection is notified to both communicating devices as part of the process of establishing the voice connection and terminating the initial control connection.

Once the main interaction between mobile telephone and messaging centre is no longer required, the call is ceased and all further control passed back to the initial control path 170.

The roles of the various components of a communications network implementing the present invention according to the preferred embodiment are as follows:

The Remote Access Server: provides the mobile telephone with access to IP connectivity. The Remote Access Server provides a route between the mobile network and thereby the mobile telephone, whether the actual communications use packet based bearers or circuit-switched bearers, and it can allocate the IP address of the client and the IP addresses of other IP facilities such as Internet Symbolic Names resolution services and it can additionally provide authentication services, thereby only allowing authorised access.

The Short Messaging Service Centre: provides the message switching service for the short message service offered by GSM and many other networks, whereby short messages of some 150bytes can be sent from mobile telephone to mobile telephone or a defined fixed location reliably, as SMS has an optional store and forward mechanism which is used when either the mobile telephone or fixed device is not available or has no current spare capacity to receive the SMS.

The USSDC: provides a message switching service for USSD similar to that provided by the SMSC for SMS. However, the nature of USSD, with less reliability and no store and forward capability, permits simpler and faster delivery than SMS.

The messaging centre: contains a service environment containing a voice message store as a minimum and could encompass an entire unified messaging environment in more complex embodiments. To access this messaging centre from mobile telephones and other suitable calling devices requires communication access which conventionally would be telephone or ISDN lines etc, but in this case would be using packetized encoded voices as described earlier using IP as the communications protocol to deliver the packets of voice. The messaging centre also needs the control functions of a WAP communications environment 150, or conventional TCP/IP and HTML environment, which provides the means of notification and management of the users interaction with the messaging services provided, for example notifying which messages you have, whether old, or new and who from, etc, options to listen, move to next, delete, save, forward, and many other basic controls, and optionally more complex functions such as conversion from/to voice to/from text, fax etc. Although the WAP gateway for protocol handling of SMSC is shown as an integral component of the Messaging Centre, it could equally be external of the Messaging Centre.

Figure 2 also illustrates that the network infrastructure is unchanged when using the existing InterWorking Function to convert between the network specific protocols and those more commonly expected outside of a mobile network, for example PSTN or ISDN. The modified IWF function 200 additionally would contain the IP voice packetization protocols necessary for provision of more general voice services to mobile thereby removing the conventional voice encoding and replacing it with just IP voice like services over a number of bearer service options, e.g. conventional circuit switched or high speed general radio packet bearers

According to preferred embodiments of the invention, calling devices such as the Messaging Centre thus provide means for performing an initial notification (via, for example, SMS or USSD), followed by the caller and called devices setting up a call in which voice data traffic and associated ongoing control information can be handled by the one call. The user thus requires only one call and the network has more SMS and USSD capacity available for other users and services. From the user's perspective, the cost of use of the service is only that of the voice call during the period when the enhanced voice services are being used. For the operator, the improved performance provided by this invention increases customer satisfaction while the reduction in SMS or USSD traffic during such periods, although reducing revenue if the bearer services are being paid for rather than being amortised into the service costs, generates better response for those still using the service within the network cell. This benefits the other users and potentially avoids the costs of adding additional channels or dedicated SMS, USSD capacity which would increase costs without necessarily revenue.

Recent developments in voice and Internet technology have brought about facilities for managing the delivery of voice based information using Internet Protocol based communications. This involves sending voice information in digital form in discrete packets rather than using the traditional circuit switched protocols of the telephone networks. Any of the emerging packet based services may be used (such as GPRS being deployed by GMS, CDMA Packet, or UMTS Packet, etc).

A specific mechanism for delivery of voice data over IP (referred to hereafter as 'VoIP') was derived from the 'VoIP Forum', an effort by major equipment providers including Cisco, VocalTec, 3Com, and Netspeak to promote the use of communications protocol standard ITU-T H.323. This is becoming the standard for sending audio and video using IP on the public Internet and within intranets. The Forum also promotes the use of directory service standards so that users can locate other users and the use of touch-tone signals for automatic call distribution and voice mail.

In addition to IP, VoIP uses the real-time protocol (RTP) to help ensure that packets get delivered in a timely way. Using public networks, it is currently difficult to guarantee Quality of Service (QoS). Better service is possible with private networks managed by an enterprise or by an Internet telephony service provider (ITSP).

A technique used by at least one equipment manufacturer (Netspeak) to help ensure faster packet delivery is to ping all possible network gateway computers that have access to the public network and choose the fastest path before establishing a TCP socket connection with the other end.

Using VoIP, an enterprise positions a "VoIP device" (such as Cisco's AS5300 access server with the VoIP feature) at a gateway. The gateway receives packetized voice transmissions from users within the company and then routes them to other parts of its intranet (local area or wide area network) or, using a T-1 or E-1 interface, sends them over the public switched telephone network.

For VoIP, the voice information is sampled after suitable filtering, most commonly 8-bit (or more) samples at 8K samples/second which results in 64kbps or higher. This level of speech quality is consistent with the norms of the telecommunications industry which normally uses 8K samples/s with 8 bits/sample. The subsequent encoding in GSM brings the transmitted data rate down to below 13kbps for full rate speech, or 6.5kbps for a recently proposed coding and decoding of voice which achieves a half-transmission-rate voice service with acceptable voice quality. Framing this within IP packets of modest size, balancing the need to use small packets of perhaps 64 or 128 bytes/packet to minimise loss of speech and recovery, and allowing for overheads which would tend towards the use of larger packet results in full rate speech of approximately 16-19 kbps with packets of 64-128 bytes of data, or 8-9.5 kbps for half rate speech.

Standard full rate GSM speech coding and decoding has too high a data rate to squeeze into current 9600bps data calls. However, with increases in GSM's basic data speeds, IP encoded full rate speech can be considered. However the option of using a half rate speech codec with a 9600bps or higher data circuit will allow sufficient spare data capacity to provide low bandwidth data applications such as WAP enhanced services to coexist in the same IP communications circuit. This is made easier to achieve by the existence of natural quiet periods in speech, the presence of which can be relied upon to some extent.

As noted above, protocol H.323 has been established for the delivery of IP Voice based services. While more complex than is absolutely

necessary for IP Voice in the case of GSM or other mobiles networks using a point-to-point circuit switched call based form of connection it would be much more relevant where GSM's GPRS (General Packet Radio Service) is considered and hence this is one option for implementation of the packetized voice delivery of the present invention.

Thus, the problem is solved in the following way for the enhanced voice messaging service used above for illustration.

A mobile phone user has for some reason not been able to receive some incoming calls. This might have been for one of a number of reasons, such as: the phone was in use, turned off or not in coverage of the network. The messaging center generates a PUSH alert to the user in the form of a WML deck which is sent to the user via the WAP proxy 150 using the default bearer service available, e.g. USSD. Having received the USSD content the phone displays it via the micro-browser user interface. The user can now decide whether to defer this notification or take some action upon it. Assuming the latter, the user chooses to listen to the message. By selecting the action LISTEN TO MESSAGE, the phone could optionally send a response to the messaging service via the WAP proxy and must instruct the phone to establish a data call to the Remote Access Server gateway giving IP access to both the WAP proxy and the messaging service. Upon establishment of the IP connection via the RAS gateway, a message is sent to the messaging service to establish the new identity if the message was sent earlier, and, additionally, to send the instructions to LISTEN if the message was not sent earlier. Having set up all the identities and instructions the messaging service can now start to provide the IP Voice. At some point the user can interrupt with new instructions etc via the IP data connection to the messaging center via the WAP proxy. At the end of the session, the client can send one final message to the WAP proxy to delete the current temporary RAS identity and to resume the default connection, then disconnect the call to the RAS gateway.

To illustrate the ease with which the present invention can be implemented, a set of changes which can be made to the components involved in a WAP based service provision over GSM for implementing the invention will now be described:

The mobile telephone can use existing voice processing functions for both the transmission and reception processing (voice encoding and decoding) to achieve the necessary raw data rate supported by this method over the available connection bit-rate. The mobile telephone would additionally need:

1. Internet Protocol communications capability (e.g. UDP/IP) and PPP based data protocol connection establishment in addition to any SMS or USSD support for the WAP environment which would be required for the basic services function. This is unlikely to be an increase on functional requirement for the phone in many instances as the WAP environment will ideally use IP for some services demanding larger amounts of data, e.g. 'over the air programming', or even basic services when SMS or USSD may not be available.

2. the ability to packetize the encoded voice into IP packets for transmission via the connection. For received information there is a need to receive the packetized voice and to deliver this to the voice decoding circuits of the phone.

3. the ability to multiplex the voice and data control packets. This should be a normal IP function, but some buffering of the voice packets would ideally be provided to permit the synchronously (regularly) sampled voice to be sent plesiochronously (of a nominal rate but not synchronously) over the network.

The WAP communications functions of the messaging centre, or even a separate gateway, would not require any changes since the ability to resume connections on different bearers is already defined.

The messaging centre would require no changes to its control operation other than the support of WAP, which is typically required regardless of whether the present invention is implemented. However, the messaging centre would require the same function as the phone to encode, decode, buffer and manage the packetization of voice traffic over IP.

The WAP Proxy is provided with the capability to support identity updates/changes (this is catered for in the WAP specifications). The routing between the messaging center, RAS gateway and preferably, though less importantly, the WAP Proxy requires sufficient capacity to avoid congestion if the voice over IP protocol being used over the mobile link does not have the full capabilities of H.323 to buffer etc.

Throughout the examples above, it has been assumed that the established Internet principles of port numbers are used to identify the application, namely voice over IP, and therefore no explicit service identification bits are required. However this invention recognises that signalling bits within the IP packet conveying the encoded and packetized voice could convey other useful information (such as quiet period duration, encoding algorithms etc) in order to maximise the options and performance over available bearer capacity.

There are alternative ways in which the cost problem associated with multiple dedicated channels could have been addressed, but each alternative has its problems.

Firstly, it would be possible to use DTMF signalling within the audio band. Simple choices could certainly be managed this way but complex messages or further WML messages might be very inefficient. One potential problem is that in some networks signalling during a call, such as DTMF, is achieved using USSD with reconstruction in the network and thus no gain is made, just an increase in complexity. As indicated previously such features are network type and deployment dependent.

Secondly, the call could be split into two portions, one portion being the conventional voice call and the other being a data call purely for the conveyance of control information. At any time either the voice or data call can be utilised, each with their bespoke communication format and protocol, but only one at a time. This could reduce users' costs, but it is more complex than the present invention and places greater demands on network infrastructure resources even if the number of channels required at any one time is the same as for the present invention. Whether the user would then be charged for one or two calls is a network service provisioning choice. Features aimed at supporting such concurrent voice or data calls are emerging, but the required network features are complex and not all networks will have this capability.

The invention has been described above as using an initial notification via narrow-bandwidth bearers and than use of wide-bandwidth bearers for IP voice services and ongoing control. Alternative implementations which only use the packet based service are possible, but these are less desirable in terms of costs of bearers, battery power consumption, and network utilisation. A solution using packet based services only would require the RAS server to be terminating IP based communications or equivalent from the network infrastructure rather than using circuit-switched calls.

CLAIMS

1. A method for use in the communication of audio information between a wireless communication device and a remote communication device via a wireless communication network, the method comprising:

at one of said communication devices, providing audio information in digital form in discrete packets and providing call control information in digital form in discrete packets;

establishing a call connection between the wireless communication device and the remote communication device; and

transmitting both the discrete audio packets and the discrete call control packets via said single call connection.

2. A method according to claim 1, wherein said step of establishing a call connection includes:

sending a notification from one of said communication devices to the other one of said communication devices via a first connection;

establishing said call connection between said devices;

verifying the identities of the calling devices;

notifying each of said devices that subsequent control information is to be exchanged via said call connection; and

closing or suspending said first connection.

3. A method according to claim 2, including notifying each of said devices that use of a connection other than said single connection is to be resumed in response to closing said single call connection or ending of the communication session.

4. A method according to claim 2, wherein the content of said notification, audio packets and control packets comprise WMLScript and/or WML, or another other XML-based or HTML-based language.

5. A method according to one of claims 2 to 4, wherein said notification is transmitted via a low-bandwidth bearer service and said call connection uses a wide-bandwidth bearer service.

6. A method according to any one of the preceding claims, wherein said remote communication device is a messaging centre providing voice-based communication services, the method including the steps of:

sending via a first connection a notification from the messaging centre to the wireless communication device of the availability of a message to be delivered to the wireless communication device;

sending a notification from the wireless communication device to the messaging centre of the wireless communication device's intention to establish a connection for voice-based communication;

the wireless communication device and the messaging centre establishing said call connection;

verifying the identities of the communication devices;

notifying each of said devices that subsequent control information is to be exchanged via said established call connection; and

closing or suspending said first connection.

7. A method according to any one of the preceding claims, wherein said discrete audio packets and control packets are transmitted using the Internet Protocol over a circuit-switched call connection.

8. A method according to claim 7, wherein said audio packets and control packets are transmitted using protocol ITU-T H.323 over the Internet Protocol.

9. A method according to any one of the preceding claims, wherein said step of providing audio information in digital form in discrete packets includes encoding voice data using a coding scheme which achieves a transmission-rate of approximately 19 kbps or less.

10. A method according to claim 9, wherein the encoding using the coding scheme achieves a transmission-rate of approximately 9.5 kbps or less.

11. A wireless communication device, including:

means for encoding audio information in a digital form in discrete packets and means for generating discrete packets of digital call control information;

means for establishing a call connection via an access node of a communications network;

means for transmitting both the discrete audio packets and the discrete control packets to the local access node via said call connection;

means for receiving discrete packets containing audio information and discrete packets containing call control information from the network access node via said call connection; and

means for decoding received audio information packets and control information packets.

12. A messaging centre for providing voice-based communications services in a communications network supporting wireless communications, the network including access nodes for receiving signals from, and for transmitting signals to, wireless communications devices within their local cell, the messaging centre including:

means for encoding audio information in a digital form in discrete packets and means for generating discrete packets of digital call control information;

means for establishing a call connection between the messaging centre and a wireless communications device within the network;

means for transmitting the encoded discrete audio packets and discrete control packets to the wireless communications device via said call connection;

means for receiving discrete packets containing audio information and discrete packets containing call control information sent across the network from the wireless communications device; and

means for decoding received audio information packets and control information packets.

13. A messaging centre according to claim 12, including:

means for transmitting to a wireless communications device via a wireless communications network a first message including, within the message content, one or more menus of selectable operations; and

means responsive to receipt of a message, sent from said wireless communications device in response to the first message, for establishing said call connection for provision of a selected operation.

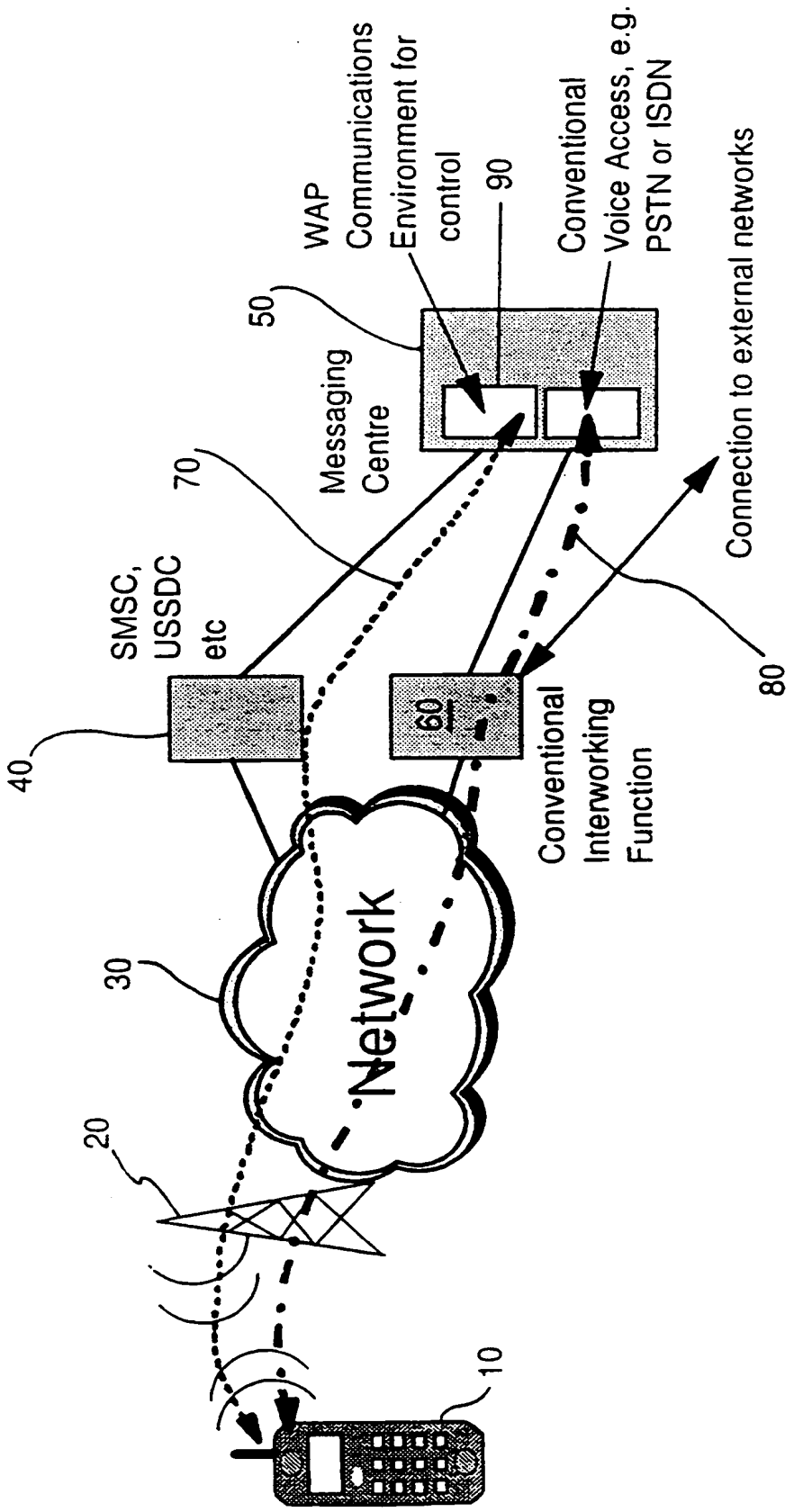
WAP
Communications
Environment for
control
90
Conventional
Voice Access, e.g.
PSTN or ISDN
50
Messaging
Centre
70
SMSC,
USSDC
etc
Connection to external networks
40
60
80
Conventional
Interworking
Function
Network
30
20
10

Figure 1

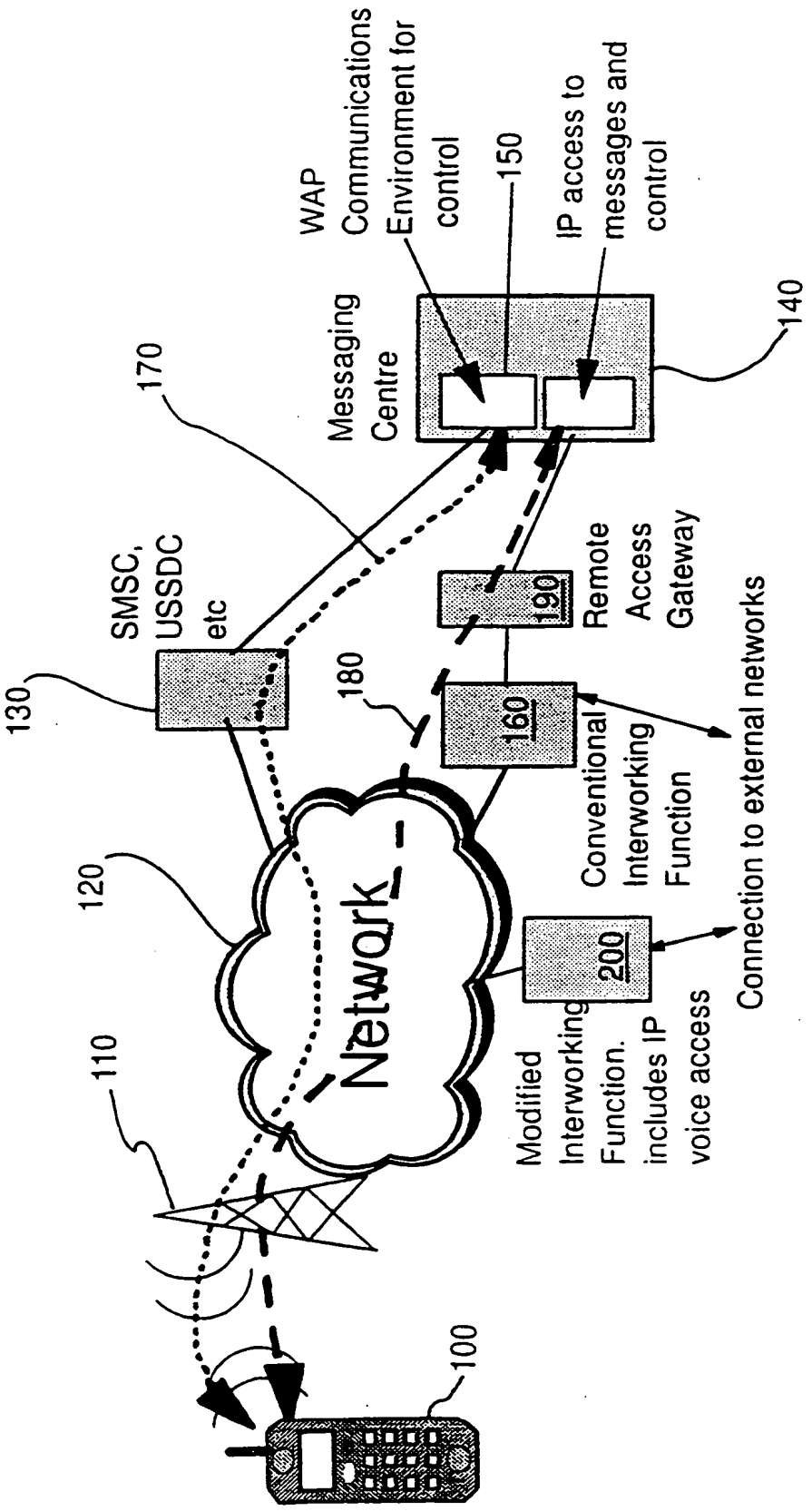

Figure 2

INTERNATIONAL SEARCH REPORT

Intl. Jonal Application No
PCT/GB 99/04304

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04Q7/22 H04L12/56

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04Q H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5 596 573 A (BERTLAND STAFFAN) 21 January 1997 (1997-01-21) column 3, line 46 - line 66 column 4, line 21 - line 67	1-3,5,6, 11,12
X	WO 96 09708 A (JORMALAINEN JANNE ;NOKIA MOBILE PHONES LTD (FI); HAEMAELAEINEN JAR) 28 March 1996 (1996-03-28) page 2, line 19 - line 24 page 6, line 13 - line 14 page 7, line 32 - line 38 page 8, line 13 - line 14 page 8, line 33 - line 36	1,11

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
22 March 2000	28/03/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Rothlübbers, C

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INTERNATIONAL SEARCH REPORT

Int. ional Application No
PCT/GB 99/04304

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	ERLANDSON C ET AL: "WAP-the wireless application protocol" ERICSSON REVIEW,SE,ERICSSON. STOCKHOLM, vol. 75, no. 4, 1998, pages 150-153-153, XP002101097 ISSN: 0014-0171 ---	4,7
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INTERNATIONAL SEARCH REPORT

Information on patent family members

Inte. ...ional Application No
PCT/GB 99/04304

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WO 9609708 A	28-03-1996	FI 944352 A	21-03-1996
		AU 3389695 A	09-04-1996